Physical Modelling as a Proposed Framework for the Conception the Design and the Implementation of Sound Transformations

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PHYSICAL MODELLING AS A PROPOSED FRAMEWORK FOR THE CONCEPTION, THE DESIGN AND THE IMPLEMENTATION OF SOUND TRANSFORMATIONS

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ABSTRACT

Since the early days of electronic and computer music, a variety of methods dedicated to the processing of musical signals have been proposed, designed and developed by musicians and sound engineers. Every underlying technique and technology that has been used for the realization of the audio processing systems offered different types of sound transformations and proposed new ways of control. The advent of signal processing by digital computers stimulated deeply the researchers in this domain for the conception of new audio effects. On the other hand, modelling and digital simulation formalisms have been principally used for the merely imitation and emulation of older sound processing systems. The aim of this article is to propose an approach for the conception, the design and the implementation of digital audio effects based on physical modelling.

1. INTRODUCTION

The history of musical sound modification has been written by innovative musicians and pioneering researchers in the domain of audio and acoustics. Mechanical, acoustical, electromechanical, electromagnetic, electronic and digital systems were developed and used for musical sound transformations. Schroeder and Logan demonstrated the first computer simulation of reverberation in 1961 using a simple digital signal processing algorithm based on comb filters and all-pass networks [18]. Since that first numerical approach, many new digital audio effects algorithms were designed using digital signal processing techniques.

From a technical point of view, digital audio effects are digital systems that modify audio signals [24]. These transformations are made according to some control parameters that the algorithm permits and deliver output sounds. The control in a wide sense signifies all the possible methods available to the user for accessing the various parameters of the system (GUIs, abstract algorithms, physical models, gestural interfaces, sound features).

According to Trevor Wishart a sound transformation is the process to change a sound were sound is referred as a sound material or a sound source [22]. Even if he does not precise whether this process is signal processing procedure, the idea behind sound

transformation is a certain mathematical manipulation of a representation of sound. Generally, the sound transformation in the context of digital and analogue computation has always been treated from the scope of signal manipulation.

Physical modelling, even if it is the most active field in digital sound synthesis the last twenty years, has rarely been proposed as method for the design and the conception of new digital audio effects algorithms [13]. Modelling techniques have been used mainly to emulate analogue audio signal processing systems as the Moog filter [20][10] and modulation effects [11]. Julius Smith has also used the digital waveguide scheme for artificial hall reverberation [19] or to emulate spring reverbs [1].

We propose a different approach of musical sound transformation based on the physical simulation of vibrating structures: in a fist step the audio signal is transmitted and propagated through a properly designed virtual viscoelastic system; then a set of mechanical manipulations of this system is taking place which accordingly transforms the sound. This procedure offers a purely material nature in the sound modification. It is not the signal that is manipulated but the matter.

In the next section, some primitive concepts concerning sound transformation that stimulated this research will be exposed. Then, the physical modelling formalism that has been used (CORDIS-ANIMA) will be presented and compared with the signal processing approaches that underlie almost every digital audio effect. Some particular characteristics of this modelling scheme will be analysed within the framework of audio processing. Finally two examples will be given.

2. SOUND TRANSFORMATIONS

The idea of sound transformation is inherent to electroacoustic music. It refers to the process of transforming a sound into another with different quality [21]. A more musical oriented definition describes sound transformations as "the processing of sound to highlight some attribute intended to become part of the musical discourse within a compositional strategy" (*Glossary* of EARS web site [23]).

It is evident that the field of sound transformation is lying between science and art. It could be seen strictly scientifically as a branch of signal processing applied to sound signals. However the goal is neither obvious nor clear as in audio engineering. Trevor Wishart whose contribution on sound transformation is invaluable says

characteristically [22]: "…In particular the goal of the process set in motion may not be known or even (with complex signals) easily predictable beforehand. In fact, as musicians, we do not need to "know" completely what we are doing (!!). The success of our effort will be judged by what we hear…"

The mathematics and the rigorous scientific background are more important for the analysis of algorithms (when this is possible). Digital reverberators were invented through experimentation and they were so difficult to analyze. The analysis started taking place many years after the first design. Much of the mathematics concerns also the impact of the finite precision effects. Fortunately nowadays by the use of the floating point representation of numbers and also by sufficient long word sizes we managed to minimize those problems.

The composition centered on sound transformations as the *musique concrète* is not based on the note/instrument conception of music. Further more is interesting to approach it as an exploration of the sound modification universe and avoid the paradigm of the intelligent instrument designer/creative user. The composer may be responsible for both the design and the control of his algorithms.

Hence the development of procedures which manipulate the sound is considered as an important part of the musical work. It is essential to mention that the processing algorithms and the source sounds are strongly related. As Wishart says, the sonic art of sound manipulation is not similar to the arranging techniques used in studios. The procedures do not work with all sounds. Both of them must be properly selected to achieve the desired musical goal.

Nowadays the sound transformation tends to be a synonym to audio signal manipulation by analogue (analogue technology is mostly used for the distortion effect) or digital computations. This section will recall some classifications of audio effects and will propose a new one which is not necessary linked with the concept of signal processing.

2.1. Proposed classifications of sound transformations

Verfaille summarizes several classifications of audio effects [21]. As he points out these classifications are neither exhaustive, nor mutually exclusive.

- *Classification based on underlying techniques*: Mechanics/Acoustics, Electromechanics, Electromagnetics, Analogue Electronics, Digital Electronics. In [24] digital audio effects are organized more precisely in filters, delays, modulators/demodulators, nonlinear processing, spatial effects, time-segment processing, time-frequency processing, source-filter processing, spectral processing, time and frequency warping. A sub-classification can be based on the domain of application (time, frequency, time-frequency).

- *Classification based on the type of control*: Wave generators, automation, gestural control, adaptive/sound-defined control.
- *Classification based on Perceptual Attributes*: The effects may modify the pitch, the dynamics, the time, the space and the timbre.
- *Interdisciplinary Classification*: This classification links the pre-described ones from low level to high level features (digital implementation > processing domain> control type > perceptual attributes > semantic descriptors).

2.2. A new proposition for the classification of sound transformations

The research on timbral development from on texture to another is evident in the 20th century history of electronic and computer music. During the 20s Varèse has already started searching new sound qualities working with natural musical instruments only [16]. The early works of Iannis Xenakis are excellent examples of instrumental sound transformations. These transformations influenced a lot and motivated Trevor Wishart to start his investigations on audio effects [14]. These types of transformations are based on the mechanical manipulation of the sound propagation medium and on the excitation mechanism.

The invention of the commercial gramophone record offered a conversion of time information into spatial information. However this technique was not only used for storing sound information. Soon after composers began to experiment with the recording medium and with the process of the sound reproduction. Darius Milhaud carried several experiments investigating vocal transformations during the period 1922 to 1927 [16]. An excellent and more contemporary example of musical creation based on the manipulation of the recording support is the Disc Jockey (DJ). The whole performance is focused on the direct manipulation of the records: playing inverse, playing at different speeds, "scratching", playing with many and different type of heads, scratching with a sharp tool the surface are some techniques used by experimental Djs.

Optical recording has also given an interesting support and encouraged musical experimentations. Pfenninger in 1932 modified recorded sounds by altering their shapes on the optical soundtrack. The introduction of the magnetic tape recorder on the studios after the Second World War gave new promises on sound transformation. Once more the creative process is based on the manipulation of the support. The enhanced possibilities of tape gave birth to Musique Concrète in 1948. Even though the magnetic type systems did not permit physical modifications of the visible on the eye waveform patterns, their editing and rewriting capabilities were significantly important to musicians for musical expression.

Analogue and digital technology offered a different type of sound treatment. It was neither the sound propagation medium nor the recording support that it was manipulated but a proper mathematical

representation of sound. Analogue signal processing techniques have been used since the late 19 century with the invention of the Telharmonium. Most of the widely known audio effects like the *phazer*, the *wahwah*, the *distortion*, the *chorus* were created with analogue signal processing techniques and implemented with electronic circuits [2]. Digital signal processing continued the same idea and offered a more convenient and general framework for the conception, the design and the implementation of digital audio effects.

Summarizing all the above we may classify the sound processing techniques in three general categories:

- *Propagation Medium Processing:* A mechanical manipulation is taking place on the vibrating structure that the sound propagates. The acoustical musical instruments may be classified in this category. The signal to be treated is the excitation mechanism. The resonator and its manipulation is the audio effect. The reverberation chamber or the reverberation plate or spring is classified in this category.
- *Recording Medium Processing:* A physical (mechanical or electromagnetic) manipulation is taking place on the support where the sound is recorded. Segmenting and rearranging the support, scratching it, altering the shapes that sound takes are several techniques to modify the musical signal.
- *Information Processing:* The musical signal is seen as a mathematical signal. A set of mathematical operations transform it either in the continuous time domain (analogue signal processing) or in the discrete time domain (digital signal processing).

3. CONCEPTION, DESIGN AND IMPLEMENTATION OF SOUND TRANSFORMATION

3.1. Digital Signal Processing Approach

For the digital signal processing approach, the audio effect which is a discrete-time system is considered as an abstract mathematical operator that transforms the input sound sequence into another sequence. The input sequence is a coded representation of the sound signal. It could be a *time-domain representation*, a *time-frequency representation*, or a *parametric representation*.

Most of the digital audio effects are conceived and designed on the time-domain. One of the most intuitive ways to create sound transformations is to cut the input streams, replay them and re-assemble them in different ways. All these may be done down to sample-accuracy. Filters, delay functions, reverberation algorithms are other examples of sound transformations that may be realized in the time-domain by elementary mathematical operators as multipliers, adders and delay lines.

Time-Frequency processing permits to work with a sound signal from both frequency and time view points simultaneously. In 1946 Dennis Gabor first introduced the time-frequency representation of a signal. Each point in this time-frequency representation corresponds both to a limited interval of time and a limited interval of frequency. In general with time-frequency methods we project the signal (time representation) onto a set of basis functions to determine their respective correlations which give the transform coefficient. If these transforms are discrete the time-frequency plane takes the form of a grid or a matrix of coefficients. In the fixed resolution case as in the Phase Vocoder, the bandwidths of the basis functions/analysis grains are the same for all frequencies and they have the same length. This gives constant time-frequency resolution. In the multiresolution case as in the Wavelet Transform, the analyzing grains are of different lengths and their bandwidth is not constant. In this scenario the wider subbands give better time resolution and vice versa.

Parametric or Signal-Model processing of audio signals concerns algorithms based on sound synthesis models: the signal to be modified is first modeled and then the model parameters are modified to achieve the desired transformation. All the valuable knowledge in sound synthesis can be applied in sound transformation. Hence as Risset states [17], it would be of great interest to develop automatic analysis procedures that could, starting with a given sound, identify the parameter values of a given sound model that yield a more or less faithful imitation of sound. This is a really hard problem for certain synthesis models. The most useful and widely known sound synthesis models are the additive synthesis, the subtractive or source-filter synthesis, the frequency modulation and the wave-shaping or non-linear distortion.

The block diagrams offer a convenient structural representation of the computational algorithm of a digital audio effect. This kind of representation uses interconnected basic building blocks as adders, multipliers and unit delays. The interconnections may be in cascade, in parallel or in feedback. The "alphabet" of the composer or the DSP engineer is these basic blocks that treat the time-domain, time-frequency or parametric data (figure 1). Musical sensibilities, engineering skills, and the formalism of scientific methods are the important competences for a meaningful sound transformation. Its conception and design is based on *"Signal Thinking"* i.e. direct manipulation of the input information. Less often, a transformation can be made by affecting the parameters of the analysis/synthesis procedures.

The design of audio effects turned the signal processing into an art. The medium and the partition of the composer could be considered as the network. He does not use notes and durations but signals and operations on signals. Some times it is even possible to figure out the sound treatment by inspection of the topology. This type of the procedure decomposition in elementary signal processing blocks is very well suited for sound signal transformation. The synthesis of the desirable DSP networks is based mainly on experimentation. That does not exclude the mathematically based design (a good example is the design of filters). Thus empirical cut and try methods

when integrated with scientific approaches lead to very interesting results and artistic products.
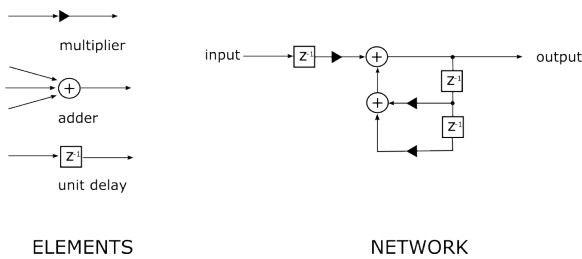

Figure 1. Digital Signal Processing basic block elements and a simple second order network

These networks offer the information needed to implement the sound transformation. They are the first step in the hardware and software implementation of the digital audio signal processing system. Computer music languages as Max MSP and Pure Data or in a more elementary level Simulink of Matlab technical computing language, work with this paradigm of information processing using block networks.

3.2. Physical Modelling Approach

Physical modelling algorithms are widely used in digital sound synthesis nowadays. Mechanical and acoustical systems governed by physical laws are modelled using several mathematical formalisms and simulated with the use of numerical techniques and digital computers. We develop a sound processing technique based on the *Propagation Medium Processing* paradigm.

In this approach, the audio effect is actually a model of a physical object. Moreover a model of gesture completes the "alphabet" which the transformation will be built from. The laws of Newtonian Mechanics, the mental image we have for the surrounded physical world, the audio and visual simulation that digital technology may offer, provide the basic tools for our research. No signal operations are used, only physical manipulation of the mater.

The initial concept behind any audio effect design within the physical modelling approach is what is referred in Physics as forced oscillations. The basic steps are simple: we design our vibrating structure, we set it in oscillation with our input signal, we re-assemble it with other structures dynamically and finally we interact with it by applying physical gestures. In this last step, we are able to establish a physical interaction between the musician and the audio effect which has virtual material substance. This is feasible by the use of suitable ergotic interfaces. It is remarkable that in this type of control there is no mapping between gesture and sound since no representation is involved in this situation, but only physical processes. Since physical models enable an intuitive representation of the action we perform with real objects we can easily imagine several physical gestures to play with our vibrating structure: damping, pulling, pushing, etc. We must note

that this is still true for non real-time simulations and without the use of force feedback gestural interfaces but by designing models that simulate the physical gesture. The deferred-time simulation permits to design accurate and valid models of the control gesture with a precision that is not possible in the real-time situations.

Among the various physical modelling methodologies that have been introduced and proposed in the last thirty years we will adopt the CORDIS-ANIMA (CA) formalism to accomplish our research. Its modularity, its capability to integrate and simulate the instrumental relationship with or without gestural interfaces and the essential possibility that offers for audio-visual simulation via the GENESIS simulation environment makes it the most proper formalism that fulfils our needs.
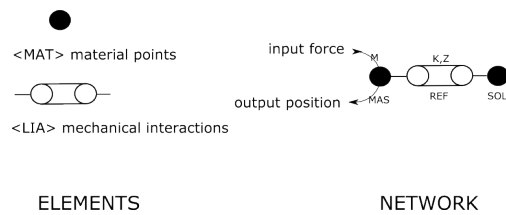

Figure 2. CORDIS-ANIMA Physical Modeling elements and simple network (elementary oscillator)

In CORDIS-ANIMA formalism [4] (further explanation can be found in the next section) a physical object is modelled as a modular assembly of elementary mechanical components. We represent the model as a plane topological network whose nodes are the punctual matter elements <MAT> and links are the physical interaction elements <LIA> (figure 2).

The simulation space used for sound and musical applications is limited to one dimension. Forces and displacements are projected on a single axis, perpendicular to the network plane. Consequently the geometrical distance between two <MAT> elements is reduced to their relative distance on the vibration axis [12].

A model is fully described by its topology-network, its values of inertia $M[i]$, elasticity $K[i]$ and viscosity $Z[i]$ and its initial conditions $X0[i]$ and $V0[i]$ (i index stands for the <MAT> number). The elasticity and viscosity may be non-linear. In the general case, within this formalism, a nonlinear interaction is represented by finite state automaton, which dynamically computes at each sample of time, the values $K[i]$ and $Z[i]$ of the stiffness and of the viscosity terms that define a physical state. Each state change is carried out by a set of conditions on the physical variables forces, displacements and velocities. More precisely in the current version of Genesis we use, the non-linear interaction is defined by a number of points on the $(F_k, \Delta x)$ and $(F_z, \Delta v)$ planes and by a simple conditional (to position) viscoelastic interaction. In GENESIS all CA models are designed graphically directly on the

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workbench as networks using a simplified representation of CA networks enriched with colours.

This highly modular representation gives the possibility to design a model based on intuition. As the basic building elements have actually a strong physical counterpart they remain pertinent to human senses and create a very realistic mental model. Therefore the design phase allows a purely physical approach carried out by "Physical Thinking". Castagne points out [9] "…Models are more easily internalized as representations of real objects than with more mathematical or signal processing physical modelling techniques…" Furthermore it is very often possible to guess and predict the general behaviour of a model by examining its network without the use of mathematical analysis tools.

As in the digital signal processing approach of musical sound transformation, the composer works with networks. It is very interesting to compare those two methodologies. The signal processing world permits to deal with signals directly: we hear them, we observe them, we analyze them and we manipulate them. In an analogous way the physical modelling universe of CA permits to deal with physical objects directly: we hear them, we observe their dynamic behaviour during the simulation, we analyse them and we manipulate them. We could similarly say that the CA formalism turned the physical modelling into an art. The composer design easily his models by using the network representation, he excites them with the input sounds, he studies how they react during the visual simulation, he rearranges and manipulates them physically using real or simulated control gestures.

This type of control is based on the "Physical Instrumental Interaction". In this control scheme we don't affect the parameters of the model -even though it is possible and previewed within the CA system- but we apply forces to the <MAT> elements of the model using <LIA> elements like in reality. It is straightforward that this type of control is totally physical and energetic coherent.

CA networks in contrast with signal processing block diagrams do not give direct information about their functional and algorithmic structure. The used algorithms and realization structures beyond the models do not appear in this representation. Other representations of CA formalism give more precise information about the calculation and the applied implementation. All these questions concerning the representation and the formalism of CA are examined in another article to published by the authors. However the algorithms have already been published [5]. So if they are accompanied by the synthesised network and the parameters, the final algorithm is directly obtained.

Even without the precise algorithms, those networks offer a simple diagrammatic explanation of the procedure. This is interesting because on the one hand they depict efficiently the basic idea of the treatment but on the other hand they leave the opportunity for several designs and realisations. We must note that CA as a

simulation language has been designed in order to offer an optimal implementation correspondence to its modularity.

4. A BRIEF VIEW IN CORDIS-ANIMA AND GENESIS

Before presenting two general models designed for musical sound transformation it is important to take a brief look on the CORDIS-ANIMA language. This step is significant in order to understand them.

CORDIS-ANIMA is a real-time mass-interaction physical modeling and simulation system [4]. This lumped parameter formalism starts from the quantization of physical matter and time. It allows designing and simulating virtual objects that can be seen, heard and handled. Moreover it offers a complete internal description of the system in all the levels of the simulation, as it provides all the essential information for its internal physical structure. All the parts of the physical objects are modeled without the use of "black boxes" The simulated objects are composed only from two types of elements, called modules:

- <MAT> modules represent punctual material elements. The most used is the MAS module, which simulates an ideal inertia. <MAT> modules are elementary subsystems and can be characterized in terms of their input/output relationships.
- <LIA> modules represent physical interactions between pairs of <MAT> modules. Available interactions are based on linear or nonlinear elasticity and friction. <LIA> modules are elementary subsystems and can be characterized in terms of their input/output relationships.

Thus, CA models are combinations of several <MAT> and <LIA> modules based on some simple construction rules.

Position and force are the two fundamental variables upon which CA modules operate. At each sample a <LIA> computes two opposite forces according to the relative distance and/or velocity of the two <MAT> it links while a <MAT> computes its position according to the forces it receives from the <LIA> modules it is linked with. The algorithms can be found on [5].

The CA system is designed and intends to simulate the "Physical Instrumental Interaction". In this interaction, the "ergotic function" [3][6][7] which is what allows in a direct way to act on the physical instrument and to feel it by the haptic sense, plays an essential role. This is what permits to perform the gesture in an expressive way and then to produce and even transform expressively sounds. In the digital sound synthesis or transformation, the "ergotic function" can be supported by specific force-feedback gestural transducers [8][15].

GENESIS [9] is a graphical environment for musical creation based on CA. The user builds CA models at an elementary level, since models are created by direct graphical manipulation and connection of individual modules on a virtual workbench. A number of higher-level tools are available for editing multiple parameters

at the same time, generating large structures, visualizing models during simulation, etc. GENESIS implements ten types of modules. While CA does not specify the dimensionality of the modules, GENESIS' simulation space is one-dimensional. <MAT> modules can only move in a perpendicular direction to the workbench, and distances and velocities are computed along this axis. For convenience, graphical manipulations take place in the 2D-space of the workbench, but the position of the modules on this plane have absolutely no consequence on the simulation: the workbench representation is only topological.

The normal set of GENESIS' building blocks is composed of:

- Linear modules: ideal mass (MAS), fixed point (SOL), second-order damped oscillator (CEL), elasticity (RES), friction (FRO), elasticity and friction combined (REF);

- Nonlinear interactions: the BUT and the LNL;

- Output modules: the SOX and the SOF, which respectively record a position and a force signal.

- Input modlules: the ENX and the ENF, which read an input file and respectively translate its data into a time-changing position (ENX) or force (ENF).

5. SOME EXAMPLES

At this point the essay will focus on two general models, which can be used for sound transformation purposes. For both of them, the input sound is considered as force applied to certain mass in the CA linear topology. The output is the position of a mass in the same topology. This force input excites the system and accordingly sets it in vibration. At this moment several other simple physical objects that are not part of the principal vibrating structure where the force waves are transmitted, interact with the structure. This type of interaction which is not linear gives the quality of the audio effect.

All linear dynamical systems in terms of signal processing are filters. Hence the linear CORDIS-ANIMA networks where all the <LIA> elements are terminated by a <MAT> element can be characterized in the general case as multiple input-multiple output linear systems. Additional, the modal analysis for the discrete time CA models, informs us that a large and interesting set of CA networks may be seen as a linear combination of second order IIR digital filters called resonators. This research does not concern the design of linear systems according to some desired specifications such as frequency response, phase response, group delay e.t.c. All these questions were examined on the same time by the authors and they are about to be published in the near future. Linear models, time variant and time invariant give effects like frequency selective filters, comb filters, chorus, flangers, phasers and reverberators.

5.1. The Tapped String Model

This model is made of 3 basic elements (fig. 3): *i)* The linear *String Model*, which is a sequence of <MAT> and <LIA> elements and approximates a real string of a musical instrument. The *String Model* is excited by an input file (force input). In our case we used approximately 20 MAS, 2 SOL and 21 REF elements. The string isolated gives a comb filter able to be tuned to a desired frequency. *ii)* The *Finger Model,* which is a heavy MAS linked to a certain MAS in the *String Model* ($M_{FingerModel} >> M_{StringModel}$). This type of nonlinear link is a conditional to position viscoelastic interaction that we call BUT. The *Finger Model* simulates the finger of a guitarist, violinist e.t.c. *iii)* The *Fingerboard Model, which is simply* a SOL linked to the MAS of the *Finger Model* by a BUT. The parameters of the BUT are calculated such us to obtain critical dumping. This model helps to stop the movement of the *Finger Model.* It simulates the fret of a string instrument. We used many *Finger Models* to give an articulation to the sound transformation. The transient state characteristics make the transformation richer than a simple time varying comb filter. This signal processing algorithm is open to experimentation. By using several sets of parameter values we may get different results. This effect works better when using sharp sounds which have a fast attack like percussive sounds.
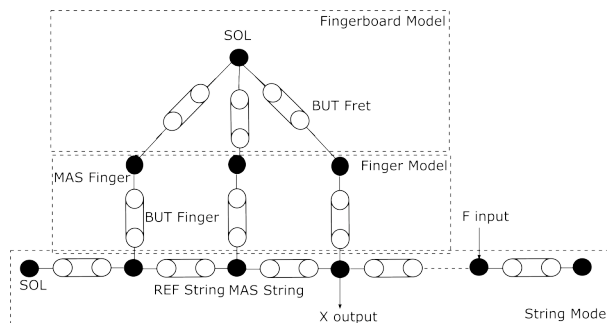

Figure 3. CORDIS-ANIMA network for the *Tapped String Model*

We could tune up the *String Model* to a pre-defined set of frequencies. However this is not a trivial task. Optimization algorithms like the Newton method or other algebraic techniques like the Cauer procedure have been used for this purpose. These results could be the subject of another article.

Figure 3 illustrates the CA network of our algorithm. The diagram depicts all the necessary information to design the effect.

5.2. The Physical Distortion Model

This simple model gives a variety of sound transformations. The basic idea is to enclose a mass driven by the input sound file into a space where the inner walls (which may be not totally rigid) are moving. The most basic version of this model is constituted by 2 elements (figure 4): *i)* The *Oscillator Model,* which is

simple linear oscillator attached to a stable point. It oscillates at subaudio frequencies. We use two of them placed in symmetrical positions. An oscillator is a combination of a REF, a MAS and a SOL module. *ii)* The *Enclosed Mass Model* (MAS module) which interacts with the oscillators by two nonlinear viscoelastic links conditioned to position (BUT). The relation of the masses of the two elements is $M_{OscillatorModel} > M_{EnclosedMassModel}$ and is decisive to the performance of the model.

This transformation often is similar to distortion. If we substitute the moving mass with an oscillator with flat frequency response we approach more this effect. The oscillator is following accurately the input signal and when it reaches the obstacles modelled by the BUT modules it stops its movement. This causes a kind of clipping on the signal. The characteristics of the BUT link characterize this clipping.
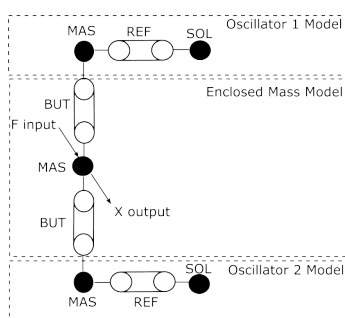

Figure 4. CORDIS-ANIMA network for the *Physical Distortion Model*

In the case of the free mass the effect is more complicated. We still obtain a kind of clipping accompanied by a more dramatical phenomenon. During the silent parts of the input sequence, the mass continues to oscillate between the two edges and thus produces a triangular signal. Its frequency is determined by the distance of the two masses (MAS module of the *Oscillator Model*) and the dissipation of the system.

6. CONCLUSION AND FUTURE WORKS

This article presented an attempt to approach sound transformations by the application of physical modelling techniques. The aim was rather to define a general framework for the concept, the design and the implementation of digital audio effects than to present list of designed models. This physical approach enhanced by the modern computer simulation techniques can give a new orientation and interesting possibilities on the research concerning the musical sound processing.

Physical modelling offers a totally different language and an "alphabet" which the transformations are built from. We have chosen the CORDIS-ANIMA scheme for its modularity, its capability to integrate and simulate the instrumental relationship with or without

gestural interfaces and the essential possibility that offers for audio-visual simulation via the GENESIS simulation environment. This decision does not exclude other formalisms like the digital waveguides to be used for similar purposes. On the contrary it would be really interesting to examine what each formalism proposes regarding the design of new effects.

Several effect prototypes have already been designed and will soon be used in electroacoustic compositions based on sound transformations. At the same time a more scientific work is oriented around the analysis of these algorithms and the systematic exploration of this physical approach.

Concluding this essay, as Trevor Wishart has once mentioned:

"Making a good transformation is like writing a tune…There are no rules" [14].

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